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PATENT APPLICATION

of

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for a

SYSTEM FOR LIMITING LOUDSPEAKER DISPLACEMENT

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SYSTEM FOR LIMITING LOUDSPEAKER DISPLACEMENT

Field of the Invention

This invention generally relates to electro-acoustical transducers (loudspeakers), and more specifically to signal processing for limiting a vibration displacement of a coil-diaphragm assembly in said loudspeakers.

Background of the Invention

The Problem formulation

A signal driving a loudspeaker must remain below a certain limit. If the signal is too high, the loudspeaker will generate nonlinear distortions or will be irreparably damaged. One cause of this nonlinear distortion or damage is an excess vibration displacement of a diaphragm-coil assembly of the loudspeaker. To prevent nonlinear distortion or damage, this displacement must be limited.

Displacement limiting can be implemented by continuously monitoring the displacement by a suitable vibration sensor, and attenuating the input signal if the monitored displacement is larger than the known safe limit. This approach is generally unpractical due to the expensive equipment required for measuring the vibration displacement. Thus some type of a predictive, model-based approach is needed.

Prior art solutions

The prior art of the displacement limiting can be put into three categories:

1. Variable cut-off frequency filters driven by displacement predictors.
2. Feedback loop attenuators.
3. Multi-frequency band dynamic range controllers.

The prior art in the first category has the longest history. The first such system was disclosed in US Patent No. 4,113,983, "Input Filtering Apparatus for

Loudspeakers", by P. F. Steel. Further refinements were disclosed in US Patent No. 4,327,250, "Dynamic Speaker Equalizer", by D. R. von Recklinghausen and in US Patent No. 5,481,617, "Loudspeaker Arrangement with Frequency Dependent Amplitude Regulations" by E. Bjerre. The essence of the prior art in the first category, utilizing a variable high pass filter with a feedback control for said displacement limiting, is shown in Figure 1a.

In this category of loudspeaker protection systems (as shown in Figure 1a), a high-pass filter **12** of a signal processor **10** filters the input electro-acoustical signal **22**. Then a filtered output signal **24** of said high-pass filter **12** is sent to a loudspeaker **20** (typically, through a power amplifier **18**) and also fed to a feedback displacement predictor block **14**. If the value of the displacement exceeds some predefined threshold value, a feedback displacement prediction signal **26** from the block **14** indicated that and a cut-off frequency of the high-pass filter **12** is increased based on the feedback frequency parameter signal **28** provided to the high-pass filter **12** by a feedback parameter calculator **16** in response to said feedback displacement prediction signal **26.** By increasing the cut-off frequency of the high-pass filter**12**, lower frequencies in the input signal, which generally are the cause of the excess displacement, are attenuated, and the excess displacement is thereby prevented.

The prior art in the first category has several difficulties. The high-pass filter **12** and the feedback displacement predictor block **14** have finite reaction times; these finite reaction times prevent the displacement predictor block **14** from reacting with sufficient speed to fast transients. Bjerre presented a solution to this problem in US Patent No. 5,481,617 at the expense of significantly complicating the implementation of the displacement limiting system. An additional problem comes from the fact that the acoustic response of the loudspeaker naturally has a high-pass response characteristic: adding an additional high-pass filter in the signal chain in the signal processor **10** increases the order of the low-frequency roll-off. This can be corrected by adding to the signal processor a low-frequency boosting filter after the high-pass filter, as was disclosed by Steel in US Patent No. 4,113,983. However, this further complicates the implementation of the signal processing.

Prior art in the second category was disclosed in US Patent No. 5,577,126, "Overload Protection Circuit for Transducers", by W. Klippel. Figure 1b shows the essence of a loudspeaker protection system describing this category. The output of the displacement predictor is fed-back into the input signal, according to a feedback parameter κ, calculated by a threshold calculator. This category of the vibration displacement protection is simpler than the first category system described above, in that it does not require a separate high-pass filter.

Prior art in the second category can be effective for the vibration displacement limiting. However, the feedback loop has an irregular behaviour around a threshold value, due to a modification of the loudspeaker's Q-factor, and an amplification at low frequencies. These effects can cause subjectively objectionable artifacts. In the above-mentioned US Patent No. 5,577,126, Klippel describes one solution to this problem: the attenuation of the signal processor is somewhat better behaved if the pure feedback signal path **16** is differentiated, as shown in Figure 3 of US Patent No. 5,577,126. However, this causes significant and unnecessary attenuation of the higher frequency band. Therefore, signals that are not responsible for the excess displacement are likely to be attenuated, degrading the performance of the loudspeaker system.

Prior art in the third category was disclosed in WO Patent Application No. PCT/EP00/05962 (International Publication Number WO 01/03466 A2), "Loudspeaker Protection System Having Frequency Band Selective Audio Power Control", by R. Aarts. Figure 1c shows the essence of the third category loudspeaker protection system. The input signal is divided into N frequency bands by a bank of band-pass filters. The signal level in the n^{th} frequency band is modified by a variable gain g_n. The signals in the N frequency bands are summed together, and sent to the power amplifier and loudspeaker. An information processor monitors the signal level in each frequency band, as modified by each of the variable gains $g_1, g_2, \ldots g_n$. The information processor modifies the variable gains $g_1, g_2, \ldots g_n$ in such a way as to prevent the excess displacement in the loudspeaker. The advantage of the third category approach is that the signal is attenuated in only that frequency band that is likely to cause the excess loudspeaker diaphragm-coil displacement. The remaining

frequency bands are unaffected, thereby minimizing the effects of the displacement limiting on the complete audio signal.

The disadvantage of the third category displacement limiter is that there are no formal rules describing how the information processor should operate. Specifically, no formal methods are available for describing how the information processor should modify the gains g_n so as to prevent the output signal from driving the loudspeaker's diaphragm-coil assembly to the excess displacement. The information processor can only be designed and tuned heuristically, i.e., by a trial-and-error. This generally leads to a long development time and an unpredictable performance.

Summary of the Invention

The object of the present invention is to provide a novel method of signal processing for limiting a vibration displacement of a coil-diaphragm assembly in electro-acoustical transducers (loudspeakers).

According to a first aspect of the invention, a method for limiting a vibration displacement of an electro-acoustical transducer comprises the steps of: providing an input electro-acoustical signal to a low frequency shelving and notch filter and to a displacement predictor block; generating a displacement prediction signal by said displacement predictor block based on a predetermined criterion in response to said input electro-acoustical signal and providing said displacement prediction signal to a parameter calculator; and generating a parameter signal by said parameter calculator in response to said displacement prediction signal and providing said parameter signal to said low frequency shelving and notch filter for generating an output signal and further providing said output signal to said electro-acoustical transducer thus limiting said vibration displacement.

According further to the first aspect of the invention, the electro-acoustical transducer may be a loudspeaker.

Further according to the first aspect of the invention, the low frequency shelving and notch filter may be a second order filter with a z-domain transfer function given by

$$H_c(z) = \sigma_c \frac{1 + b_{1\cdot c} z^{-1} + b_{2\cdot c} z^{-2}}{1 + a_{1\cdot t} z^{-1} + a_{2\cdot t} z^{-2}},$$

wherein σ_c is a characteristic sensitivity of the low frequency shelving and notch filter, $b_{1\cdot c}$ and $b_{2\cdot c}$ are feedforward coefficients defining target zero locations, and $a_{1\cdot t}$ and $a_{2\cdot t}$ are feedback coefficients defining target pole locations. Further, said parameter signal may include said characteristic sensitivity σ_c and said feedback coefficients $a_{1\cdot t}$ and $a_{1\cdot t}$.

Still further according to the first aspect of the invention, the method may further comprise the step of: generating said output signal by the low frequency shelving and notch filter. Further, the method may further comprise the step of: providing the output signal to said electro-acoustical transducer. Yet further, the output signal may be amplified using a power amplifier prior to providing said output signal to said electro-acoustical transducer.

According further to the first aspect of the invention, the displacement prediction signal may be provided to a peak detector of the parameter calculator. Still further, after the step of generating the displacement prediction signal, the method may further comprise the step of: generating a peak displacement prediction signal by the peak detector and providing said peak displacement prediction signal to a shelving frequency calculator of the parameter calculator. Yet still further, the method may further comprise the step of: generating a shelving frequency signal by the shelving frequency calculator based on a predetermined criterion and providing said shelving frequency signal to a sensitivity and coefficient calculator of the parameter calculator for generating, based on said shelving frequency signal, the parameter signal.

According still further to the first aspect of the invention, the input electro-acoustical signal may be a digital signal.

According further still to the first aspect of the invention, said low frequency shelving and notch filter may be a second order filter with an s-domain transfer function given by

$$H_c(s) = \frac{s^2 + s\omega_c/Q_c + \omega_c^2}{s^2 + s\omega_t/Q_t + \omega_t^2},$$

wherein Q_c is a coefficient corresponding to a Q-factor of the electro-acoustical transducer, ω_c is a resonance frequency of the electro-acoustical transducer mounted in an enclosure, Q_t is a coefficient corresponding to a target equalized Q-factor, ω_t is a target equalized cut-off frequency. Still further, Q_c may be equal to $1/\sqrt{2}$, when the electro-acoustical transducer is critically damped. Yet further, Q_c may be a finite number larger than $1/\sqrt{2}$, when the electro-acoustical transducer is under-damped.

According to a second aspect of the invention, a computer program product comprising: a computer readable storage structure embodying computer program code thereon for execution by a computer processor with said computer program code, characterized in that it includes instructions for performing the steps of the first aspect of the invention indicated as being performed by the displacement predictor block or by the parameter calculator or by both the displacement predictor block and the parameter calculator.

According to a third aspect of the invention, a signal processor for limiting a vibration displacement of an electro-acoustical transducer comprises: a low frequency shelving and notch filter, responsive to an input electro-acoustical signal and to a parameter signal, for providing an output signal to said loudspeaker thus limiting said vibration displacement of said electro-acoustical transducer; a displacement predictor block, responsive to said input electro-acoustical signal, for providing a displacement prediction signal; and a parameter calculator, responsive to said displacement prediction signal, for providing the parameter signal.

According further to the third aspect of the invention, the parameter calculator block may comprise: a peak detector, responsive to the displacement prediction signal, for providing a peak displacement prediction signal; a shelving frequency calculator, responsive to the peak displacement prediction signal; for providing a shelving frequency signal; and a sensitivity and coefficient calculator, responsive to said shelving frequency signal, for providing the parameter signal. Further still, said

low frequency shelving and notch filter may be a second order digital filter with a z-domain transfer function given by

$$H_c(z) = \sigma_c \frac{1 + b_{1\cdot c} z^{-1} + b_{2\cdot c} z^{-2}}{1 + a_{1\cdot t} z^{-1} + a_{2\cdot t} z^{-2}},$$

wherein σ_c is a characteristic sensitivity of the low frequency shelving and notch filter, $b_{1\cdot c}$ and $b_{2\cdot c}$ are feedforward coefficients defining target zero locations, and $a_{1\cdot t}$ and $a_{2\cdot t}$ are feedback coefficients defining target pole locations. Yet further, said parameter signal may include said characteristic sensitivity σ_c and said feedback coefficients $a_{1\cdot t}$ and $a_{1\cdot t}$.

Further according to the third aspect of the invention, the output signal may be provided to said electro-acoustical transducer or said the output signal is amplified using a power amplifier prior to providing said output signal to said electro-acoustical transducer.

Still further according to the third aspect of the invention, the input electro-acoustical signal may be a digital signal.

According further to the third aspect of the invention, the low frequency shelving and notch filter may be a second order filter with an s-domain transfer function given by

$$H_c(s) = \frac{s^2 + s\omega_c/Q_c + \omega_c^2}{s^2 + s\omega_t/Q_t + \omega_t^2},$$

wherein Q_c is a coefficient corresponding to a Q-factor of the electro-acoustical transducer, ω_c is a resonance frequency of the electro-acoustical transducer mounted in an enclosure, Q_t is a coefficient corresponding to a target equalized Q-factor, ω_t is a target equalized cut-off frequency. Further, Q_c may be equal to $1/\sqrt{2}$, when the electro-acoustical transducer is critically damped. Yet still further, Q_c may be a finite number larger than $1/\sqrt{2}$, when the electro-acoustical transducer is under-damped.

According still further to the third aspect of the invention, the electro-acoustical transducer may be a loudspeaker.

Brief Description of the Drawings

For a better understanding of the nature and objects of the present invention, reference is made to the following detailed description taken in conjunction with the following drawings, in which:

Figures 1a, 1b and 1c show examples of a signal processor and a loudspeaker arrangement for a first, second and third category signal processing systems for a loudspeaker protection (vibration displacement limiting), respectively, according to the prior art.

Figures 2a shows an example of a signal processor with a loudspeaker arrangement utilizing a variable low-frequency shelving and notch filter driven by a feedforward control using a displacement predictor block, according to the present invention.

Figures 2b shows an example of a parameter calculator used in the example of Figure 2a, according to the present invention.

Figure 3 shows an example of response curves of a low-frequency shelving and notch filter (without a notch and Q_c=0.707) for a critically damped loudspeaker, according to the present invention.

Figure 4a and 4b show examples of displacement response curves for a loudspeaker which is critically damped and under-damped, respectively, by utilizing a low-frequency shelving and notch filter of Figure 3, according to the present invention.

Figure 5a shows an example of response curves of a low-frequency shelving and notch filter (with a notch and Q_c=6.4) for an under-damped loudspeaker, according to the present invention.

Figure 5b shows an example of displacement response curves for a loudspeaker which is under-damped by utilizing a low-frequency shelving and notch filter of Figure 5a, according to the present invention.

Figure 6 is a flow chart demonstrating a performance of a signal processor with a loudspeaker arrangement utilizing a variable low-frequency shelving and notch

filter driven by a feedforward control using a displacement predictor block, according to the present invention.

Best Mode for Carrying Out the Invention

The present invention provides a novel method for signal processing limiting and controlling a vibration displacement of a coil-diaphragm assembly in electro-acoustical transducers (loudspeakers). The electro-acoustical transducers are devices for converting an electrical or digital audio signal into an acoustical signal. For example, the invention relates specifically to a moving coil of the loudspeakers.

The problems of the prior art methods described above for the displacement limiting is solved by starting with the first category approach, and making the following modifications:

- Replacing the variable high-pass filter **12** (see Figure 1a) with a variable low-frequency shelving and notch (LFSN) filter;
- Using a feedforward instead of a feedback control of the filter **12** by the displacement predictor block;
- Employing a digital implementation;
- Approximating the exact formulas for calculating required coefficients by finite polynomial series.

According to the present invention, a signal processor with the above characteristics or a combination of some of these characteristics provides a straightforward and efficient system for said displacement limiting. Large signals that can drive the loudspeaker into an excess displacement are attenuated at low frequencies. Higher-frequency signals that do not overdrive the loudspeaker can be simultaneously reproduced unaffected. The behaviour of the limiting system can be known from its base operating parameters, and can therefore be tuned based on the known properties of the loudspeaker.

Figure 2 shows one example among others of a signal processor with a loudspeaker arrangement utilizing a low-frequency shelving and notch (LFSN) filter **11**

driven by a feedforward control using a displacement predictor block **14a** for limiting a vibration displacement of an electro-acoustical transducer (loudspeaker) **20**, according to the present invention. The limiting of the vibration displacement is achieved by modifying a transfer function of the LFSN filter **11** based on the output of the displacement predictor block **14a**.

As in Figure 1a, the LFSN filter **11** of a signal processor **10a** filters the input electro-acoustical signal **22**. Said input electro-acoustical signal **22** can be a digital signal, according to the present invention. Then a filtered output signal **24a** of said high-pass filter **11** is sent to a loudspeaker **20** (typically, through a power amplifier **18**). But, according to the present invention, the input electro-acoustical signal **22** is also fed to a displacement predictor block **14a**. If the value of the vibration displacement exceeds a predefined threshold value (that is a predetermined criterion), a displacement prediction signal **26a** from the block **14a** is generated and provided to the parameter calculator **16** which generates a parameter signal **28a** in response to that signal **26a** and then said parameter signal **28a** is provided to the LFSN filter **11**. Based on said parameter signal **28a**, the transfer function of said LFSN filter **11** is modified appropriately and the output signal **24a** of said LFSN filter **11** has the vibration displacement component attenuated based on said predetermined criterion.

The LFSN filter **11** attenuates only low frequencies, which are the dominant sources of a large vibration displacement. The diaphragm-coil displacement can be predicted from the input signal **22** by the displacement predictor block **14a** implemented as a digital filter. Generally, the required order of said digital filter is twice that of the number of mechanical degrees of freedom in the loudspeaker **20**. The output of this filter is the instantaneous displacement of the diaphragm-coil assembly of the loudspeaker **20**. The performance of the displacement predictor block **14a** is known in the art and is, e.g., equivalent to the performance of the part 9 shown in Figure 2 of US Patent No. 4,327,250, "Dynamic Speaker Equalizer", by D. R. von Recklinghausen. Detailed description of the parameter calculator **1a** is shown in an example of Figure 2b and discussed in detail later in the text.

The LFSN filter **11** can be designed, according to the present invention, as a second-order filter with an s-domain transfer function given by

$$H_c(s)=\frac{s^2+s\omega_c/Q_c+\omega_c^2}{s^2+s\omega_t/Q_t+\omega_t^2} \quad (1),$$

wherein Q_c is a coefficient corresponding to a Q-factor (of the loudspeaker **20**), ω_c is a resonance frequency of a loudspeaker **20** mounted in a cabinet (enclosure), in rad/s, Q_t is a coefficient corresponding to a target equalized Q-factor, ω_t is a target equalized cut-off frequency (shelving frequency), in rad/s. The magnitude of the frequency response of the filter **11,** a low-frequency gain, equals to ω_c^2/ω_t^2. Typical gain curves for this low-frequency shelving and notch filter **11** with $Q_c = Q_t = 1/\sqrt{2}$ (the loudspeaker **20** is critically damped and the LFSN filter **11** does not have a notch) are shown in Figure 3 for five values of ω_t^2/ω_c^2 ratio. The ability of the LFSN filter **11** to limit the displacement is made clear in Figure 4a.

Figure 4a shows an example among others of displacement response curves for the loudspeaker **20**, which is critically damped by utilizing the LFSN filter **11** of Figure 3, according to the present invention. As the value of ω_t is increased, the displacement response is attenuated as seen in Figure 4a. In the low frequency limit, the amount of attenuation varies as ω_t^2. The mathematical detail behind this is discussed below. These displacement response curves are for a "critically damped" loudspeaker, i.e., one tuned to a Butterworth alignment ($Q_c = Q_t = 1/\sqrt{2}$).

Inexpensive loudspeakers often have an under-damped response, i.e., having values of Q_c and Q_t greater than $1/\sqrt{2}$. Figure 4b shows an example of displacement response curves for the loudspeaker **20** which is under-damped, by utilizing the LFSN filter **11** of Figure 3, according to the present invention. The higher Q_c and Q_t values of the loudspeaker **20** make the relationship between the reduction in the displacement response and the increase in ω_t less straightforward, particularly near the resonance frequency ω_c. To solve this problem, the value of Q_c may be "artificially" decreased. This is done by setting the value of Q_c in Equation 1 to the value of $Q_c \approx 6.4$ (instead of $1/\sqrt{2}$). Figure 5a shows an example among others of response curves of the low-

frequency shelving and notch filter **11** (with a notch at ω_c by setting Q_c=6.4) for an under-damped loudspeaker **20**, according to the present invention. As can be seen from Figure 5a, the resulting response has a notch at the resonance frequency ω_c, which comes from setting the numerator Q-factor in Equation 1 to a value higher than $1/\sqrt{2}$. For this reason, it the filter **11** is referred to as the low frequency shelving and notch (LFSN) filter.

The effect of the LFSN filter **11** on the displacement response of the under-damped loudspeaker **20** is demonstrated in Figure 5b. The broken line shows the loudspeaker's displacement response without the LFSN filter.

The transfer function describing the ratio of the vibration displacement to the input signal **22** is a product of the LFSN filter **11** response (transfer function) and the loudspeaker **20** displacement response. This is an equalized displacement response in the s-domain given by

$$\begin{aligned} H_{DP\cdot E}(s) &= H_c(s) X_{m\cdot v_c}(s) \\ &= \frac{\phi_0}{m_t R_{eb}} \frac{s^2 + s\omega_c/Q_c + \omega_c^2}{s^2 + s\omega_t/Q_t + \omega_t^2} \frac{1}{s^2 + s\omega_c/Q_c + \omega_c^2} \end{aligned} \quad (2),$$

which reduces to

$$H_{DP\cdot E}(s) = \frac{\phi_0}{m_t R_{eb}} \frac{1}{s^2 + s\omega_t/Q_t + \omega_t^2} \quad (3),$$

wherein ϕ_0 is a loudspeaker's transduction coefficient (B· l factor), R_{eb} is a DC-resistance of the voice coil of the loudspeaker **20** and m_t is a total moving mass.

The reduction of Equation 2 to Equation 3 is an important result for operating the displacement predictor block **14a** of Figure 2a. The input to the displacement predictor block **14a** is the input signal **22**, not the output signal **24a** from the LFSN filter **11** (as in the prior art, see Figure 1a). Thus the displacement predictor block **14a** must account for the effect of the LFSN filter **11**. It would at first seem that the displacement predictor would need to account for the second-order system described by the loudspeaker displacement response $X_{m\cdot v_c}(s)$ and the second order LFSN filter **11**, resulting in a fourth-order system altogether. However, the reduction of Equation

2 to the single second-order transfer function described by Equation 3 shows that the displacement predictor block **14a** needs only be a second-order system.

The same reduction can be made for the *z*-domain transfer function describing a digital processing implementation of the equalized displacement response. The product between the *z*-domain transfer functions of the digital processing version of the LFSN filter **11** and a digital model of the loudspeaker **20** displacement is given by

$$H_{DP\cdot E}(z)=\sigma_c\sigma_{x\cdot v_c}\frac{1+b_{1\cdot c}z^{-1}+b_{2\cdot c}z^{-2}}{1+a_{1\cdot t}z^{-1}+a_{2\cdot t}z^{-2}}\frac{z^{-1}}{1+a_{1\cdot c}z^{-1}+a_{2\cdot c}z^{-2}} \quad (4),$$

wherein σ_c is a characteristic sensitivity of the LFSN filter, $\sigma_{x\cdot v_c}$ is a characteristic sensitivity of the digital displacement predictor block **14a**, $b_{1\cdot c}$ and $b_{2\cdot c}$ are feedforward coefficients defining the target zero locations, $a_{1\cdot t}$ and $a_{2\cdot t}$ are feedback coefficients defining the target pole locations and $a_{1\cdot c}$ and $a_{2\cdot c}$ are feedback coefficients defining the loudspeaker's pole locations.

It is noted that the coefficients $b_{1\cdot c}$ and $b_{2\cdot c}$ can have the same values as $a_{1\cdot c}$ and $a_{2\cdot c}$, respectively. Therefore Equation 4 reduces to

$$H_{DP\cdot E}(z)=\sigma_c\sigma_{x\cdot v_c}\frac{z^{-1}}{1+a_{1\cdot t}z^{-1}+a_{2\cdot t}z^{-2}} \quad (5).$$

The Equation 5 can be written with a single characteristic sensitivity by defining

$$\sigma_{dp\_m}=\sigma_c\sigma_{x\cdot v_c} \quad (6),$$

wherein σ_{dp_m} is the metrically correct characteristic sensitivity, given by

$$\sigma_{dp\_m}=\frac{a_g\phi_0}{R_{eb}k_t}\left(1+a_{1\cdot c}+a_{2\cdot c}\right)\frac{1-a_{1\cdot t}+a_{2\cdot t}}{1-b_{1\cdot c}+b_{2\cdot c}} \quad (7),$$

wherein a_g is a gain of the power amplifier **18a** and D/A converter (not shown in Figure 2a but used in a case of the digital implementation) and k_t is a total stiffness of the loudspeaker **20** suspension (loudspeaker's suspension stiffness) including acoustic loading from any enclosure.

The LFSN filter **11** achieves limiting the vibration displacement by increasing the frequency ω_t. As shown in Figures 3 and 5a, increasing this frequency ω_t reduces the gain at lower frequencies, and leaves it unchanged at higher frequencies. This provides the desired limiting effect, by changing the displacement response as shown in Figures 4a and 5b.

The displacement-limiting algorithm is shown in more detail in Figure 2b. A peak detector **16a-1**, in response to the displacement prediction signal **26a** from the displacement predictor block **14a**, provides a peak displacement prediction signal **21** to a shelving frequency calculator **16a-2.** The peak detector provides an absolute value of the displacement. It also provides a limited release time (decay rate) for the displacement estimate.

As discussed above, at low frequencies, the gain of the filter varies according to the square of the shelving frequency. Due to the nature of the displacement response of the loudspeaker **20**, it is assumed that the signals that are responsible for the excess displacement are at the low frequencies. With this assumption, the required shelving frequency is calculated from the excess displacement as follows:

$$
\begin{aligned}
&\mathit{if}\ \left(x_{pn}[n] > x_{lm}\right) \\
&\qquad f_r = f_t \sqrt{1 + \frac{x_{pn}[n] - x_{lm}}{x_{lm}}} \qquad (8), \\
&\mathit{else} \\
&\qquad f_r = f_t
\end{aligned}
$$

wherein f_r is a shelving frequency required to limit the displacement, f_t is a target cut-off frequency, x_{lm} and $x_{pn}[n]$ is a displacement predicted by the displacement predictor block **14a** and normalized to a maximum possible displacement x_{mp}.

The maximum possible displacement x_{mp} can be determined from an analysis of the displacement predictor block **20**. It can be calculated as

$$x_{mp} = \frac{g_{RX}\phi_0 F(Q_c)}{k_t R_{eb}} \qquad (8a),$$

wherein g_{RX} is a maximum possible voltage that the D/A and power-amplifier (the D/A conversion is used for the digital implementation) can create, and $F(Q_c)$ is a function of the loudspeaker's Q-factor, given by

$$F(Q_e)=\begin{cases} 1 & Q_c \le 1/\sqrt{2} \\ \dfrac{1}{\sqrt{\dfrac{1}{Q_c^2}-\dfrac{1}{4Q_c^4}}} & Q_c > 1/\sqrt{2} \end{cases} \qquad (8b).$$

The peak value is determined according to

$$\begin{aligned} & if\ \left(|x_{in}[n]| > x_{pn}[n-1]\right) \\ & \qquad x_{pn}[n] = |x_{in}[n]| \\ & else \\ & \qquad x_{pn}[n] = t_r\ x_{pn}[n-1] \end{aligned} \qquad (8c),$$

wherein $x_{in}[n]$ is an instantaneous unity-normalized predicted displacement, $x_{pn}[n]$ is a peak-value of the unity-normalized predicted displacement, and t_r is a release time constant. The release time constant t_r is calculated from the specified release rate d in dB/s, according to

$$t_r = 10^{-d/20F_s} \qquad (8d),$$

wherein F_s is a sample rate.

The required shelving frequency f_r is given by the algorithm of Equation 8. If the predicted displacement is above the displacement limit (according to a predetermined criterion), this required shelving frequency is increased from the target shelving frequency f_t according to the first expression of Equation 8. Otherwise (if the predicted displacement is below said limit), the required shelving frequency remains the target shelving frequency (see Equation 8). If the required shelving frequency changes, new values for the coefficients $a_{1\cdot t}$, $a_{2\cdot t}$, and σ_c need to be calculated by a sensitivity and coefficient calculator **16a-3**, thus providing said parameter signal **28a** to the variable LFSN filter **11.** In theory, these parameters could

be calculated by formulas for digital filter alignments. However, these methods are generally unsuitable for a real-time, fixed-point calculation. Methods for calculating these coefficients with polynomial approximations suitable for the fixed-point calculation are presented below.

An initial simplification can be made for the f_r calculation using Equation 8 by defining x_{lmg}, the inverse of the scaled displacement limit, as

$$x_{lmg} = 1/x_{lm} \qquad (9).$$

This value, x_{lmg}, is the maximum attenuation needed for the displacement limiting. Substituting x_{lmg} into the first expression of Equation 8 results in the following expression for calculating f_r:

$$f_r = f_t\sqrt{x_{lmg}}\sqrt{x_{pn}[n]} \qquad (10).$$

This value of f_r is used to calculate $\omega_{r\cdot z}$, a frequency required for the displacement limiting, in rad/s, normalized to sampling rate as follows

$$\omega_{r\cdot z} = \frac{2\pi}{F_s} f_r \qquad (11),$$

wherein F_s is a sampling rate.

Combining Equations 11 and 12 results in

$$\omega_{r\cdot z} = \frac{2\pi}{F_s} f_t\sqrt{x_{lmg}}\sqrt{x_{pn}[n]} \qquad (12).$$

By defining $\omega_{t\cdot z}$ in terms of f_t as in Equations 11 and 12 reduces it to

$$\omega_{r\cdot z} = \sqrt{\omega_{t\cdot z}^2\, x_{lmg}\, x_{pn}[n]} \qquad (13).$$

From this value of $\omega_{r\cdot z}$, new values of $a_{1\cdot r}$ and $a_{2\cdot r}$ can be calculated as follows

$$\begin{aligned} a_{1\cdot r} &= -2e^{-\omega_{r\cdot z}\zeta_r}\cos\left(\omega_{r\cdot z}\sqrt{1-\zeta_r^2}\right) \\ a_{2\cdot r} &= e^{-2\omega_{r\cdot z}\zeta_r} \end{aligned} \qquad (14),$$

wherein ζ_r is a damping ratio.

The coefficients $a_{1\cdot r}$ and $a_{2\cdot r}$ can be calculated directly from $x_{pn}[n]$, defined as a displacement normalized to the maximum possible displacement (x_{mp}) at a time sample n, by combining Equations 10 through 14. Furthermore, these coefficients can be approximated by these polynomial series in $x_{pn}[n]$.

$$\hat{a}_{1\cdot r}\left(x_{pn}[n]\right) = p_{a_1\cdot 0} + p_{a_1\cdot 1}x_{pn}[n] + p_{a_1\cdot 2}x_{pn}^2[n] + p_{a_1\cdot 3}x_{pn}^3[n] + p_{a_1\cdot 4}x_{pn}^4[n] \quad (15)$$

and

$$\hat{a}_{2\cdot r}\left(x_{pn}[n]\right) = p_{a_2\cdot 0} + p_{a_2\cdot 1}x_{pn}[n] + p_{a_2\cdot 2}x_{pn}^2[n] + p_{a_2\cdot 3}x_{pn}^3[n] + p_{a_2\cdot 4}x_{pn}^4[n] \quad (16).$$

The characteristic sensitivity σ_c can be calculated from $\hat{a}_{1\cdot r}$ and $\hat{a}_{2\cdot r}$ according to

$$\sigma_c = b_d\left(1 - a_{1\cdot r} + a_{2\cdot r}\right) \quad (17),$$

wherein

$$b_d = \frac{1}{1 - b_{1\cdot c} + b_{2\cdot c}} \quad (18).$$

The variables $b_{1\cdot c}$ and $b_{2\cdot c}$ are known from the properties of the loudspeaker **20**.

As $b_{1\cdot c}$ and $b_{2\cdot c}$ change only with the loudspeaker **20** characteristics, and thus change only infrequently, it is more efficient to compute b_d, and store this in a memory for calculating σ_c. Therefore, according to the present invention, the value of b_d can to be calculated only once (and not continuously in the real-time),

The complete formulas for $a_{1\cdot r}$ and $a_{2\cdot r}$ are difficult to approximate with short polynomial series for the full range of theoretically valid values of $\omega_{r\cdot z}$ with an adequate accuracy. Potentially, the approximation accuracy can be improved by increasing the order of the polynomial series. This has not been found to be feasible, because it not only increases significantly the complexity of the calculation, it also leads to coefficients to be poorly scaled, making them unsuitable for the fixed-point calculation.

The solution to this problem is to optimize the accuracy of the polynomial coefficients which can mean that different polynomial coefficients will have to be used for different hardware and sampling rates, as the latter can be known for a given product, so such coefficients can be stored in that product's fixed ROM.

Using $x_{pn}[n]$ as the input to the polynomial approximation has an additional advantage. Since all of x_{pn}, $a_{1\cdot r}/2$, $a_{2\cdot r}$, and σ_c are limited to the range (0, 1), the values of the polynomial coefficients in the polynomial approximation will be better scaled than if, e.g., the required cut-off frequency is used as the input to the polynomial approximation Using said $x_{pn}[n]$ simplifies implementation of the polynomial approximation using a fixed-point digital signal processor. Therefore, the polynomial series can be a good approximation for calculating $a_{1\cdot r}$ and $a_{2\cdot r}$ from x_{pn}:

$$a_{1\cdot r}/2 = -e^{-\zeta_r \pi \sqrt{a_f x_{pn}}} \cos\left(\pi\sqrt{a_f x_{pn}}\sqrt{1-\zeta_r^2}\right)$$
$$a_{2\cdot r} = e^{-2\zeta_r \pi \sqrt{a_f x_{pn}}} \qquad (19),$$

wherein a_f is given by

$$a_f = \frac{1}{\pi^2}\omega_{t\cdot z}^2 x_{lmg} \qquad (20),$$

and wherein the range of possible values of x_{pn} is

$$x_{pn} \in (x_{lm}\ , 1) \qquad (21).$$

This corresponds to a possible range of values of $\omega_{r\cdot z}$ of

$$\omega_{r\cdot z} \in \left(\ \omega_{t\cdot z}\ ,\ \omega_{t\cdot z}\sqrt{x_{lmg}}\ \right) \qquad (22).$$

The Equations 7 through 22 illustrate only a few examples among many other possible scenarios for calculating a characteristic sensitivity, $a_{1\cdot r}$ and $a_{2\cdot r}$ by the parameter calculator **16a**.

Finally, Figure 6 is a flow chart demonstrating a performance of a signal processor with a loudspeaker arrangement utilizing a variable low-frequency shelving and notch filter **11** driven by a feedforward control using a displacement predictor block **14a** for limiting a vibration displacement of an electro-acoustical transducer (loudspeaker) **20**, according to the present invention.

The flow chart of Figure 3 only represents one possible scenario among many others. In a method according to the present invention, in a first step **30**, the input electro-acoustical signal **22** is received by the signal processor **10a** and provided to the LFSN filter **11** of said signal processor **10** and to the displacement predictor block

14a of said signal processor **10**. In a next step **32**, the displacement predictor block **14a** generates the displacement prediction signal **26a** and provides said signal **26a** to the peak detector **16a-1** of the parameter calculator **16a** of said signal processor **10**. In a next step **34**, the peak displacement prediction signal **23** is generated by the peak detector **16a-1** and provided to the shelving frequency calculator **16a-2** of said parameter calculator **16a**. In a next step **36**, the shelving frequency signal **23** is generated by the shelving frequency calculator **16a-2** and provided to the sensitivity and coefficient calculator **16a-3** of the parameter calculator **16a**. In a next step **38**, the parameter signal **28a** (e.g., which includes the characteristic sensitivity and polynomial coefficients) is generated by the sensitivity and coefficient calculator **16a-3** and provided it to the LFSN filter **11**. In a next step **40**, the output signal 24a is generated by the LFSN filter **11**. Finally, in a last step **42**, the output signal **24a** is provided to the power amplifier **18** and further to the loudspeaker **20**.

As explained above, the invention provides both a method and corresponding equipment consisting of various modules providing the functionality for performing the steps of the method. The modules may be implemented as hardware, or may be implemented as software or firmware for execution by a processor. In particular, in the case of firmware or software, the invention can be provided as a computer program product including a computer readable storage structure embodying computer program code, i.e., the software or firmware thereon for execution by a computer processor (e.g., provided with the displacement predictor block **14a** or with the parameter calculator **16a** or with both the displacement predictor block **14a** and the parameter calculator **16a**).